CISCO SPA232D MULTI-LINE ATA

PRODUCT OVERVIEW

The Cisco® SPA232D Multi-Line ATA is a mobility enhanced, affordable, highly reliable voice gateway for connecting an analog device to a voice-over-IP (VoIP) service provider. It can also intelligently route calls to the public switched telephone network (PSTN).

The SPA232D supports essential voice features such as caller ID, call transfer, call waiting, call forwarding, voicemail, and much more to provide a comprehensive, advanced, and highly innovative VoIP solution.

The SPA232D provides one RJ-11 FXS port and one FXO port to connect to the PSTN. The SPA232D FXS and FXO lines can be independently configured through software by the service provider or the end user. Users can take full advantage of their broadband phone service by enabling intelligent "hop-on, hop-off" applications to route local calls from mobile phones and land lines over to their VoIP service provider, and conversely.

The Cisco SPA232D also includes two 100BASE-T RJ-45 Ethernet interfaces to connect to a home or business LAN, and an Ethernet port to connect to a broadband access device. It uses international standards for voice and data networking for reliable voice operation, and it can be used in residential, small office or home office (SOHO), and business environments.

FEATURES AND BENEFITS

The Cisco SPA232D Multi-Line ATA (Figures 1 and 2) offers the following:

- High-quality feature-rich VoIP service through your broadband Internet connection
- One standard telephone port and one port for PSTN connectivity to route local calls from mobile phones and land lines over to VoIP service providers and conversely
- Highly configurable and secure remote provisioning capabilities to enable mass-scale service provider activation and deployment
- Ideal solution for residential, SOHO, and business environments

Business VoiceEdge™

COMCAST BUSINESS B4B™

Figure 1. Cisco SPA232D Multi-Line ATA

Figure 2. Ports on Cisco SPA232D

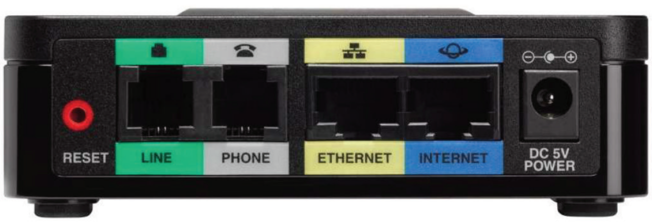

Table 1 lists additional features and benefits of the Cisco SPA232D Multi-Line ATA.

Table 1. Features and Benefits of Cisco SPA232D Multi-Line ATA

FEATURE	BENEFIT
Toll-quality voice and carrier-grade feature support	The SPA232D delivers clear, high-quality voice communication in diverse network conditions. Excellent voice quality in a demanding IP network is consistently achieved through our advanced implementation of standard voice-coding algorithms. The SPA232D is interoperable with common telephony equipment such as voicemail, PBX, interactive-voice-response (IVR) systems, and many third-party call-control systems.
Large-scale deployment and management	The SPA232D offers all the important features and capabilities with which service providers can provide customized VoIP services to their subscribers. It can be remotely provisioned, and it is softwareupgradable. A secure profile upload saves providers the time, expense, and hassle of managing and preconfiguring or reconfiguring subscriber equipment for deployment.
Outstanding security	The Cisco SPA232D supports highly secure, encryption-based methods for communication, provisioning, and servicing.
Comprehensive feature set	The standards-based Cisco SPA232D is compatible with essential Internet VoIP provider features such as caller ID, call waiting, call transfer, call forwarding, three-way conferencing, voicemail, and much more to provide a complete, affordable, and highly reliable VoIP solution.
Peace of mind	Cisco solutions deliver the solid reliability you expect from Cisco. All solution components have been rigorously tested to help ensure easy setup, interoperability, and performance.

PRODUCT SPECIFICATIONS

Table 2 gives the specifications of the Cisco SPA302D Multi-Line Handsets.

Table 2. Product Specifications

*Note: Many specifications are programmable within a defined range or list of options. Please refer to the Cisco SPA232D Administration Guide for details. The configuration profile is uploaded to the SPA232D at the time of provisioning.

DESCRIPTION	SPECIFICATION
Voice gateway	Session Initiation Protocol (SIP) v2 (RFCs 3261, 3262, 3263, and 3264) SIP proxy redundancy: Dynamic through Domain Name System (DNS) SRV record Reregistration with primary SIP proxy server SIP support in Network Address Translation (NAT) networks (including Serial Tunneling [STUN]) Highly secure (encrypted) calling with Secure Real-Time Transport Protocol (SRTP) Codec name assignment G.722 G.711 (a-law and µ-law) G.726 (32 kbps) G.729 (b and ab) Dynamic payload Adjustable audio frames per packet Dual-tone multifrequency (DTMF): In-band and out-of-band (RFC 2833) (SIP INFO)
Cordless handset* (SPA302D)	1.8-in. TFT (128 x 160 pixels), 65,000 colors, backlit with scratch-resistant lens Software upgradable over the air (SUOTA) White illuminated keypad backlight Administrative personal identification number (PIN) code support Dial keypad lock Speed dial: Eight programmable Private and shared phone books (50 records) Call history (50 records filtered by Outgoing, Incoming, and Missed) Visual Message-Waiting Indicator (VMWI) Five ringtones Call mute Call hold/resume New call (support for two active call segments) Redial Call park and unpark Intercom (handset-to-handset) Hearing Aid Compatibility (HAC) *Some features may require support by the call-control server.

Voice features	Quality of service (QoS) (Ethernet port upstream bandwidth control) Independent configurable dial plans with interdigit timers and IP dialing (per line) Call progress tone generation Jitter buffer: Adaptive Frame-loss concealment Full-duplex audio Echo cancellation (G.165 and G.168) Voice activity detection (VAD) Silence suppression Comfort Noise Generation (CNG) Attenuation and gain adjustments Flash hook timer MWI tones VMWI through frequency shift keying (FSK) Polarity control Hook flash event signaling Caller ID generation (name and number): Bellcore, DTMF, and ETSI Music-on-hold (MOH) client Streaming audio server: Up to 10 sessions MOH Call waiting and call waiting caller ID Caller ID with name and number Caller ID blocking Selective and anonymous call rejection Call forwarding: no answer, busy, and all Do not disturb Call transfer, call return, and call back on busy Three-way conference calling with local mixing Per-call authentication and associated routing Call Blocking with Toll Restriction Distinctive ringing: Calling and called number Off-hook warning tone Advanced inbound and outbound call routing Hot line and warm line calling Long silence (configurable time setting) silence threshold Disconnect tone (for example, reorder tone) Configurable ring frequency Ring validation time setting Tip and ring voltage adjustment setting Ring indication delay setting

Enhanced gateway authentication and routing features	VoIP-to-PSTN (United States) service call origination and termination PSTN-to-VoIP (United States) service call origination and termination Single- and two-stage dialing Forward calls to VoIP service: Selective, Authenticated, and All Forward calls to PSTN service: Selective, Authenticated, and All PSTN line sharing with multiple extensions Automatic PSTN fallback (loss of power or IP service to unit, with quiescence to normal operations) Advanced inbound and outbound call routing Independent configurable dial plans: Up to 8 Force PSTN disconnection Sequential dialing support **VoIP to PSTN:** • VoIP-to-PSTN gateway enable and disable • VoIP caller authentication method (None, PIN, and HTTP digest) • VoIP PIN maximum retry setting • One-stage dialing enable and disable • VoIP caller ID pattern matching • VoIP access-allowed caller list (no further authentication) • VoIP caller PIN and associated dial plan **PSTN to VoIP:** • PSTN-to-VoIP gateway enable and disable • VoIP caller authentication method (None, PIN, and HTTP digest) • Ring through to FXS enable and disable • Ring-through tone: Configurable • Caller ID (Bellcore Type 1) for VoIP service access • Caller ID enable and disable • PIN maximum retry settings • Access-allowed caller list (no further authentication) • Caller PIN and associated dial plan • Least-cost routing (through outbound VoIP: Line1 dial plan

FXO control settings	**VoIP answer-delay timer** • PSTN answer-delay timer • VoIP PIN digit timeout timer • PSTN PIN digit timeout timer • PSTN-to-VoIP call maximum-duration timer • VoIP-to PSTN call maximum-duration timer • PSTN ring-through delay timer • PSTN dialing delay timer • VoIP DLG (Dialog) refresh-interval timer • PSTN ring timeout timer **PSTN Disconnection Detection** • CPC (Calling Party Control) (removal of tip and ring voltage momentarily) • Polarity reversal • Long silence (configurable time setting) • Disconnect tone (for example, reorder tone) • Silence threshold **International Control** • FXO port impedance: Settings by country • Ring frequency: Configurable • SPA-to-PSTN and PSTN-to-SPA gain settings • Ring frequency: Maximum setting • Ring validation time setting • Tip and ring voltage adjustment setting • Ring indication delay setting • Ring timeout setting • Ring threshold • Line-in-use voltage setting
Security	Password-protected system reset to factory default Password-protected administrator and user access authority Provisioning, configuration, and authentication Secure HTTP (HTTPS) with factory-installed client certificate HTTP digest: Encrypted authentication with (MD5; RFC 1321) Up to 256-bit Advanced Encryption Standard (AES) encryption SIP Transport Layer Security (TLS) Reset button bypass (provisionable)

Data networking	MAC address (IEEE 802.3) IPv4 (RFC 791) Address Resolution Protocol (ARP) DNS-A record (RFC 1706) and SRV record (RFC 2782) Dynamic Host Configuration Protocol (DHCP) server and client (RFC 2131) DHCP client reservation DHCP Option 159 and Option 160 Point-to-Point Protocol over Ethernet (PPoE) client (RFC 2516) Internet Control Message Protocol (ICMP; RFC 792) TCP (RFC 793) User Datagram Protocol (UDP; RFC 768) Real-Time Transport Protocol (RTP; RFCs 1889 and 1890) Real-Time Control Protocol (RTCP; RFC 1889) Differentiated Services (DiffServ) (RFC 2475) and type of service (ToS; RFCs 791 and1349) VLAN tagging (IEEE 802.1p) Simple Network Time Protocol (SNTP) (RFC 2030) Upload data rate limiting: Static and automatic QoS: Voice packet prioritization over other packet types MAC address cloning Port forwarding SIP channels support for both UDP and TCP transport VPN pass-through with IP Security encapsulating security payload (IPsec ESP), Point-to-Point Tunneling Protocol (PPTP), and Layer 2 Tunneling Protocol (L2TP)
Provisioning, administration, and maintenance	Web browser administration and configuration through integral web server Telephone keypad configuration with IVR prompts Automated provisioning and upgrade through HTTPS, HTTP, and Trivial File Transfer Protocol (TFTP) TR-069 Asynchronous notification of upgrade availability using NOTIFY Nonintrusive, in-service upgrades Report generation and event logging Statistics in BYE message Syslog and debug server records: Per-line configurable web browser Configuration management: Backup and restore Support for Bonjour Support for Link Layer Discovery Protocol (LLDP) and Cisco Discovery Protocol (CDP)
Physical interfaces	One WAN 100BASE-T RJ-45 Ethernet port (IEEE 802.3) One LAN 100BASE-T RJ-45 Ethernet port (IEEE 802.3) One RJ-11 FXS phone port for analog circuit telephone device (tip and ring) One RJ-11 FXO phone port for PSTN or PBX connection Reset button
Subscriber-line interface circuit (SLIC)	Ring voltage: 40–90 Vpk configurable Ring frequency: 16–50 Hz Ring waveform: Trapezoidal or sinusoidal Maximum ringer load: Five ringer equivalence numbers (RENs) On-hook and off-hook characteristics: On-hook voltage (tip/ring): –46 ~ –56V Off-hook current: 25 mA Terminating impedance: Multiple 150 nF complex impedance Frequency response: 300–3400 Hz Return loss (600 ohm, 300–3400 Hz) >20 dB Idle channel noise: <10 dB (typical) Longitudinal balance: 60 dB (typical)

Regulatory compliance	FCC (Part 15 Class B), CE, ICES-003, A-Tick certification, Restriction of Hazardous Substances (RoHS), and UL
Power supply	DC input voltage: 5V DC at 2.0A maximum Power consumption: 5W Switching type (100–240V) automatic Power adapter: 100–240V, 50–60 Hz (26–34 VA) AC input, 6 ft.(1.8m) cord
Indicator lights and LED	Line, phone, Internet, and power
Documentation	Quick Start Guide Administration Guide: Available online Provisioning Guide: Available online
ENVIRONMENTAL	
Dimensions (H x W x D)	3.98 x 3.98 x 1.10 in. (101 x 101 x 28 mm)
Unit weight	5.40 oz. (153 g)
Operating temperature	32 to 113°F (0 to 45°C)
Storage temperature	-77 to 158°F (-25 to 70°C)
Operating humidity	10 to 90% noncondensing
Storage humidity	10 to 90% noncondensing
Package contents	Cisco SPA232D Multi-Line ATA 5V/2A power adapter 6-ft (1.83m) Ethernet cable RJ-11 telephone cable Quick Start Guide CD with documentation including license and warranty

WARRANTY INFORMATION

Cisco SPA232D is covered by a Cisco standard 1-year limited hardware warranty with return to factory replacement and 90-day limited software warranty. To download software updates, please visit: www.cisco.com/go/smallbiz.

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